

# Tour of Trixbox

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# 1. Introduce

This article is a guide to use Trixbox and ATCOM products to build a simple application on small office application. Through this article, we hope that reader can build the IP PBX solution for small enterprise

## Related Hardware and software

**Trixbox2.0:** Asterisk based Trixbox enables even the novice user to quickly set up a voice over IP phone system. Trixbox can be configured to handle a single phone line for a home user, several lines for a small office, or several T1s for a million minute a month call center.

**AX-100p:** Asterisk PCI card with One FXO port

**AX-400p:** Asterisk PCI card with four FXO/FXS interchangeable ports, four ports can be configured as FXO or FXS port individually.

**AX-110S:** FXS module of AX-400p

**AX-110X:** FXO module of AX-400p

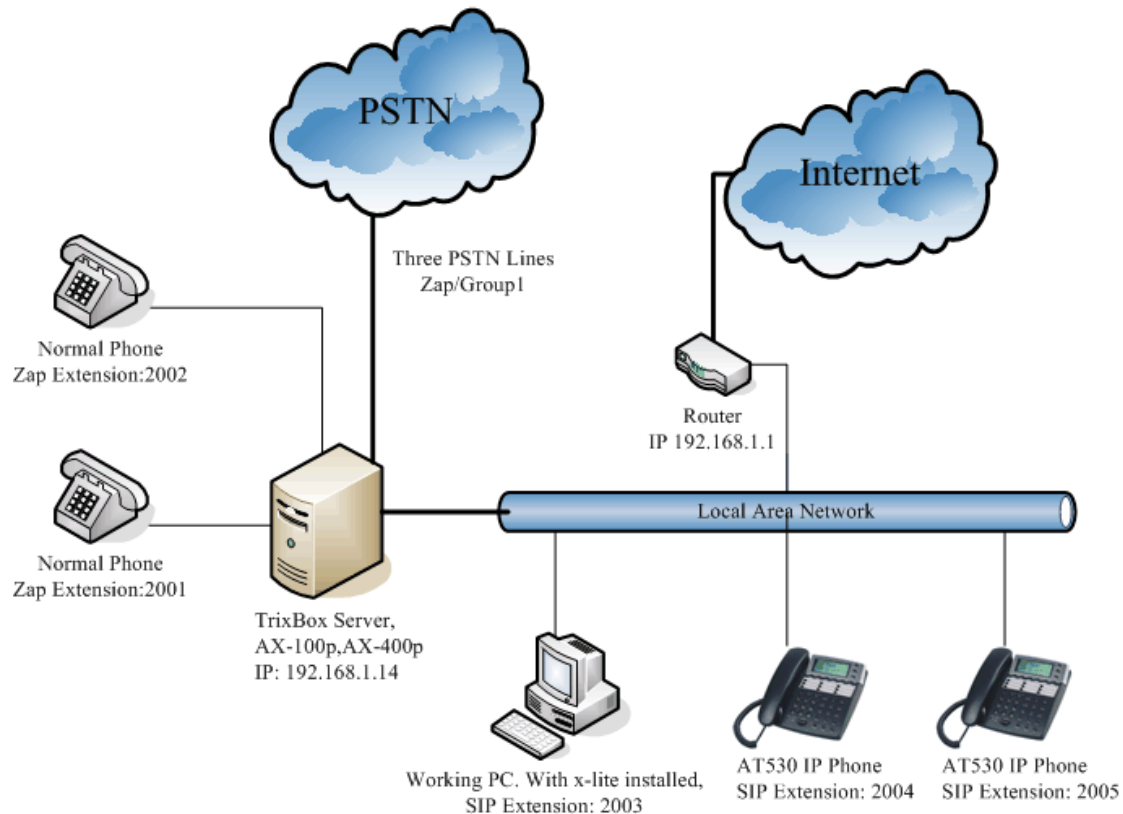
**AT-530:** Stand along IP phone, can be configured as SIP/IAX2 extensions.

**X-lite:** softphone run on the PC, can be configured as SIP extensions.

**Normal phone:** connect to AX-400p, act as ZAP extensions.

## System set up

### Simple structure for small enterprise



We use following devices in our set up:

- 1: Server with Tribox installed: we also install AX-100p and AX-400p in this server, we install two AX-110S FXS modules and two AX-110X modules, so you can connect three PSTN lines and two normal phones. So there are five analog ports on the Tribox server. one FXO port from AX-100p, two FXO port from AX-400p. two FXS ports from AX-400p.
- 2: Normal phone: connect to the AX-400p FXS ports.
- 3: Working PC, x-lite runs on this computer act as a sip extension.
- 4: AT-530 IP phone, register to Tribox to act to as SIP or IAX2 extension.

We want to provide below function in our simple application.

- 1: free internal call.
- 2: Voice Mail.
- 3: make out bound call
- 4: IVR system.

We will implement these functions step by step.

## 2. Install Trixbox and make internal calls

### Install Trixbox

The set up of Trixbox is simple. Just download the ISO file from [www.trixbox.org](http://www.trixbox.org) and record it in the CD to make an auto boot CD. PUT this installation CD and reboot your computer, then it will format your system and install the CENTOS and Trixbox server on it.

I am using Trixbox2.0 version on this article.

If you have question in install, you can refer the [www.trixbox.org](http://www.trixbox.org) for the install guide.

### Ways to Access Trixbox

There are two ways to access the Trixbox server.

- Use Web browser to open the IP of the Trixbox server to connect to the GUI of Trixbox. After open the Trixbox web interface, you need to switch the access mode from user to admin. The default username and password is **maint** and **password**. This is base configure way of Trixbox. Trixbox doesn't support IE well. You can use Firefox to configure it.
- Use the SSH tool to access the Trixbox server. in this way the use name is **root** and password is the password you input when install the Trixbox.

Most of our configure job will be done in the web interface. and we will do some advance configure via SSH tool.

### Install Trixbox modules

Trixbox doesn't install many function default, you need to install them manually. In the web interface, go to the **Asterisk -->Free PBX page --> Tools --> Module Admin**. You can see many function modules in this page. Just select all and process the install to install all these modules.

Notice: every change on the Trixbox will show a red section ask you to apply the change via click the link.

### Add extensions

At first, we need to add some extensions to make internal calls. Each extension acts as an internal number. There are many types of extensions, we will use SIP, IAX2 and ZAP extensions on this article.

### Add SIP extensions:

Sip extensions is an SIP account allows you to log on the Trixbox via an SIP terminal such as IP phone AT-530 and softphone x-lite.

Go to -->**Free PBX --> setup-->Extensions -->add generic SIP device.**

#### Add Exenions:

**User Extension:** 2003 ; Phone number of this extension

**Display Name:** Fantasy ; Caller ID

**Secret:** 2003 ; SIP Log on password

#### Enable Voicemail

**Voicemail password:** 2003 ; password of your mailbox, you need to input this password when you enter the mailbox

#### Add Extension

User Extension	<input type="text" value="2003"/>
Display Name	<input type="text" value="Fantasy"/>

#### Device Options

secret	<input type="text" value="2003"/>
dtmfmode	<input type="text" value="rfc2833"/>

#### Voicemail & Directory

Status	<input type="text" value="Enabled"/> ▼
voicemail password	<input type="text" value="2003"/>
email address	<input type="text"/>
pager email address	<input type="text"/>

Use the same method add two other extensions

**User Extension:** 2004 ; Phone number of this extension

**Display Name:** Edwin ; Caller ID

**Secret:** 2004 ; Log on password

#### Enable Voicemail

**Voicemail password:** 2004 ; password of your mailbox

**User Extension:** 2005 ; Phone number of this extension

**Display Name:** Marvin ; Caller ID

**Secret:** 2005 ; Log on password  
**Enable Voicemail**  
**Voicemail password:** 2005 ; password of your mailbox

So we have created three SIP extensions. And we can use SIP device to log on the Trixbox use these extensions.

## Register SIP device:

### 1. Register SIP softphone

Download the x-lite softphone phone from couterpath website [www.xten.com](http://www.xten.com)

After install the x-ltie, right click the panel and select the SIP Account Setting and then configure it:

The screenshot shows the 'SIP Account Setting' window of the x-lite softphone. The 'Account' tab is active. Under the 'User Details' section, the following information is entered:

Field	Value
Display Name	Fantasy
User name	2003
Password	****
Authorization user name	2003
Domain	192.168.1.129

**Display Name:** Fantasy  
**User Name:** 2003  
**Password:** 2003  
**Authorization User Name:** 2003  
**Domain:** 192.168.1.129 (IP address of your Trixbox server)

### 2. Register IP phone AT-530

- Connect the AT-530's WAN port to the switch. And it can get the ip from your router.
- Press the "sysinfo" key on AT-530 to get the IP of AT-530.
- Put the ip on the IE of your computer and you can enter the AT-530 configure page through this ip.
- Put the SIP extensions info on the AT-530 IP phones.



# IP Phone

---

## SIP[Registered] Configuration

Register Server Addr	192.168.1.129	Proxy Server Addr	
Register Server Port	5060	Proxy Server Port	
Register Username	2004	Proxy Username	
Register Password	****	Proxy Password	
Domain Realm		Local SIP Port	5060
Phone Number	2004	Register Expire Time	60 seconds
Detect Interval Time	60 seconds	User Agent	Voip Phone 1.0
Encrypt Key		Server Type	common
DTMF Mode	DTMF_RFC2833	RFC Protocol Edition	RFC3261
<input checked="" type="checkbox"/> Enable Register			

**Register Server Addr:** 192.168.1.129 IP address of Trixbox server.  
**Register Username:** 2004  
**Register Password:** 2004  
**Phone Number:** 2004

Use the same method register another at-530 to extension 2005, then you can free inbound call between these three extensions 2003~2005.

## Voice Mail Box

Voice mail box is enabled when we create the extensions. So if somebody calls you on your extension and you are unavailable, he will hear the voice mail greeting message and leave voice message.

Retrieve voice message:

Dial \*97 in your sip terminal and input the voice mail password of your extension then you will enter your voice mail box.

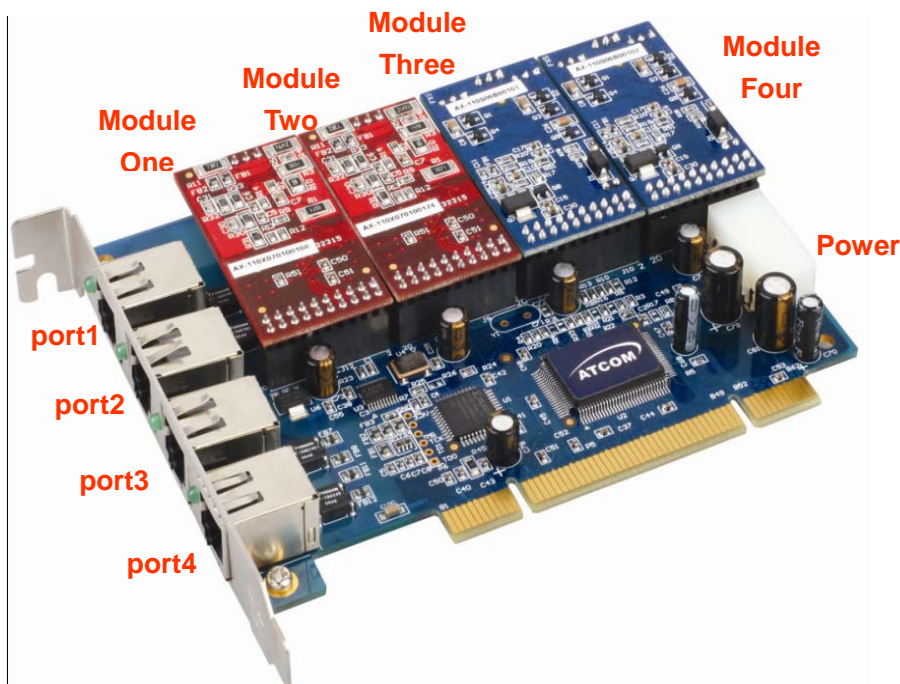
Dial \*98 in your sip terminal and you will enter the voice mail main menu, then you can input extension number and voice mail password to enter the corresponding extension's voicemail box.

### 3. Make outbound call

To make an outbound call, we need to add trunk first. There are many types of trunk, SIP trunk, Zap Trunk and so on, We use AX-100p and A-400p to make zap trunks here.

#### Install AX-100p and AX-400p to the PC (Hardware)

1. Power off Trixbox and pull out the power adapter.
2. Insert AX-100p card to the PCI slot of your PC and firm it with screw.
3. Insert AX-400p card to the PCI slot, please see below picture for the configure



**AU-110S(FXS)**



**AU-110X(FXO)**

Set up:

Insert AX-110X FXO module to module 1 and module 2, port 1 and port 2 will then be configured as FXO port.

Insert AX-110S FXS module to module 3 and module 4, port 3 and port 4 will then be configured as FXS port.

If you have insert AX-110S module to the AX-400p, because the FXS port need to provide signal to the normal phone, you also need to power the AX-400p with your PC via the power port. Otherwise AX-400p won't have sufficient voltage to drive the normal phone connected.

---

What are FXO and FXS?

**FXS** (Foreign eXchange Station) is an interface which drives a telephone or FAX

machine. FXS interfaces get phones plugged into them, delivery battery, and provide ringing. FXS interfaces are signalled with FXO signalling.

**FXO** (Foreign eXchange Office) is an interface that connect to a phone line. They supply your PBX with access to the public telephone network. FXO interfaces use FXS signalling. FXS interfaces are what allow you to hook telephones to your PBX, and FXO interfaces allow you to connect your PBX to real analog phone lines.

## Configure AX-100p and AX-400p on Trixbox

There is a command `genzaptelconf` can generate the configure file of AX-100p and AX-400p automatically.

Use the SSH tool to connect Trixbox server and run:

```
[root@asterisk1 ~]# genzaptelconf
```

Trixbox will then auto detect zaptel hardware and install AX-100p and AX-400p driver automatically.

You may see below info in this process:

```
Loading wcfxo: wcfxo: DAA mode is "FCC"
Found a Wildcard FXO: Wildcard X100P.      //Detect AX-100p card

Loading wctdm: Freshmaker version 7
Freshmaker pass register test
Module 0: Installed – AUTO FXS/DPO
Module 1: Installed – AUTO FXO (FCC mode)
Module 2: Installed – AUTO FXO (FCC mode)
Module 3: Installed – AUTO FXS/DPO
Found a Wildcard TDM: Wildcard TDM400P REV I (4 modules)
                                           //Detect AX-400p card
```

Which indicate that Trixbox detect and install AX-100p and AX-400p successfully  
And type:

```
[root@asterisk1 ~]# ztcfg -vvv
```

To see the channel state:

**Zaptel Configuration**

=====

**Channel map:**

**Channel 01: FXS Kewlstart (Default) (Slaves: 01)**

**Channel 02: FXS Kewlstart (Default) (Slaves: 02)**



## Add Zap Trunk

Use the SSH tool to open the file /etc/asterisk/zapata-auto.conf

You will see there are five channels here. Just like:

; Span 1: WCFXO/0 "Wildcard X100P Board 1" RED

signalling=fxs\_ks

; Note: this is a trunk. Create a ZAP trunk in AMP for Channel 1

context=from-zaptel

group=0

channel => 1

; Span 2: WCTDM/0 "Wildcard TDM400P REV E/F Board 1"

signalling=fxs\_ks

; Note: this is a trunk. Create a ZAP trunk in AMP for Channel 2

context=from-zaptel

group=0

channel => 2

signalling=fxs\_ks

; Note: this is a trunk. Create a ZAP trunk in AMP for Channel 3

context=from-zaptel

group=0

channel => 3

signalling=fxo\_ks

; Note: this is an extension. Create a ZAP extension in AMP for Channel 4

context=from-internal

group=1

channel => 4

signalling=fxo\_ks

; Note: this is an extension. Create a ZAP extension in AMP for Channel 5

context=from-internal

group=1

channel => 5

To use these five channels, we can use the separate via "1" "2" "3"... or use them as a group via "g0" (group=0)

Go to **FreePBX-->Set up --> Trunks** and add a new Zap trunk.

**Outbound Caller ID:** Specify your caller ID when making outbound calls. No use for FXO lines.

**Maximum Channels:** Maximum available simultaneous outbound calls.

**Outbound Dial Prefix:** 9      Trixbox will add a prefix to the number you dial and send to the trunk

**Zap Identifier:** g0      ; the three FXO ports are grouped as g0, so when you make outbound call via this trunk, Trixbox will pick up the available FXO channel in g0 automatically. The channel are grouped by Trixbox when you use “genzaptelconf”, you can see the group information in the file /etc/asterisk/zapata-auto.conf . you can also put “1” “2” or “3” here to specify the separate FXO port here

## Add ZAP Trunk

### General Settings

Outbound Caller ID:

Never Override CallerID: ☐

Maximum channels:

### Outgoing Dial Rules

Dial Rules:

Dial rules wizards:

Outbound Dial Prefix:

### Outgoing Settings

Zap Identifier (trunk name):

## Add outbound routes

Ok. We have added the trunk already. To make out bound calls, we also need to specify that which trunk that our calls should route to.

Go to the **Freepbx --> Setup--> Outbound Routes--> Add Route**

### Add Route

Route Name:	<input type="text" value="9_outside"/>
Route Password:	<input type="text"/>
PIN Set:	<input type="button" value="v"/>
Emergency Dialing:	<input type="checkbox"/>
Intra Company Route:	<input type="checkbox"/>
Dial Patterns	
<div><div>9 .</div><div>Clean &amp; Remove duplicates</div></div>	
Insert:	<div>Pick pre-defined patterns <input type="button" value="v"/></div>
Trunk Sequence	
<div>ZAP/g0 <input type="button" value="v"/></div>	
<div>Submit Changes</div>	

**Route Name:** 9\_outside

**Dial Patterns:** 9|. Trixbox will cut the first number “9” if the phone number dial begins with 9.

**Trunk Sequence:** ZAP/g0 all number accord with the above Dial Patterns will be sent to ZAP/g0 trunk.

This outbound route means, if we dial any phone number start with 9, for example 983018806, then Trixbox will cut the number first number “9” and send the number(83018806) to the g0 group (it will use the available channel of channel 1,2,3). After doing above, you can use SIP extensions to make outbound calls.

## Add zap extensions

Remember that we have two FXS modules on the AX-400p card, we can use this two fxs port to build two zap extension.

Go to the **Free PBX --> setup-->Extensions -->add generic zap device**


### Add Extension

User Extension	<input type="text" value="2001"/>
Display Name	<input type="text" value="Annie"/>

### Device Options

channel	<input type="text" value="4"/>
---------	--------------------------------

### Voicemail & Directory

Status	<input type="text" value="Enabled"/> 
voicemail password	<input type="text" value="2001"/>
email address	<input type="text"/>
pager email address	<input type="text"/>

**User Extension:** 2001  
**Display Name:** Annie  
**Channel:** 4                      the fourth channel is the FXS port  
**Enable VoiceMail**  
**VoiceMail Password:** 2001

Now connect a normal phone to the 3<sup>rd</sup> port of the AX-400p card, and pick up the phone, you will hear the dial tone and you can make calls now.

Use the same method to add another zap extension 2002

**User Extension:** 2002  
**Display Name:** Crystal  
**Channel:** 5                      the fifth channel is the FXS port  
**Enable VoiceMail**  
**VoiceMail Password:** 2002

### System Review:

So far we have:

Five internal extensions:2001~2005, calling between these extensions are free.



And we have assigned voice mail boxes to each extension. To expand the number of extensions, you can just add IP phone or softphone.

Three PSTN extensions, to make outbound calls, you need to add prefix 9 to your number when dialing.

We have finished outbound call now, the next step we will add IVR for our system.

## 4. IVR

IVR, means Interactive Voice Response. It is very important for the enterprise application. We can use the IVR to do the auto-attendant job.

### Generate the Voice file

Go to **FreePBX -->Setup --> System Recording --> Add Recording**

You can use your extension to record a voice file or Upload a voice file.

Record via extensions:

Enter an extension 2001 in the record via extension and “Go” to the next page.

### System Recordings

#### Add Recording

[Add Recording](#)[Built-in Recordings](#)

##### Step 1: Record or upload

If you wish to make and verify recordings from your phone, please enter your extension number here:

Alternatively, upload a recording in .wav format:

### System Recordings

#### Add Recording

##### Step 1: Record or upload

Using your phone, dial \*77 and speak the message you wish to record.

Alternatively, upload a recording in .wav format:

##### Step 2: Verify

After recording or uploading, dial \*99 to listen to your recording.

If you wish to re-record your message, dial \*77

##### Step 3: Name

Name this Recording:

Click "SAVE" when you are satisfied with your recording

Specify the name of this record and dial \*77 in 2001 extension and you will here indicate sound and then record your voice. When hang up and Save, now we have record a voice name “Greeting\_ATCOM” in the Trixbox

Notice: you can also upload a fair-sounding voice to the system in the “add recording” page

## Add IVR entry

Go to **FreePBX --> SetUp --> IVR--> Add IVR**

### Digital Receptionist

#### Edit Menu Unnamed

Delete Digital Receptionist Unnamed	
Change Name	<input type="text" value="Welcome"/>
Timeout	<input type="text" value="10"/>
Enable Directory	<input checked="" type="checkbox"/>
Directory Context	<input type="text" value="default"/> ▼
Enable Direct Dial	<input checked="" type="checkbox"/>
Announcement	<input type="text" value="Greeting_ATCOM"/> ▼

**Change Name:** Welcome  
**Announcement:** Greeting\_ATCOM

## Add incoming route to IVR entry

Go to **FreePBX --> SetUp --> Inbound Routes--> Add Inbound Route**

### Add Incoming Route

#### Add Incoming Route

---

DID Number:




Caller ID Number:

OR

Zaptel Channel:

#### Set Destination

---

- ☐ Recordings:  
- ☐ Core:  
- ☒ IVR:  
- ☐ Custom App:

**Zaptel Channel :**            **1**  
**Recordings:**            **Greeting\_ATCOM**

Then all PSTN incoming to channel 1 will be routed to our IVR system and they can dial any internal extension when listen the IVR

## 5. Make outbound calls via SIP Service

We have added zap trunk in the before chapter. But the rate for international call on the PSTN line is expensive. I want to more lower rate for our international call. So I apply an voipbuster account ( they have a low international rate) and after adding a SIP trunk, we can use our extension to make calls to via the voipbuster service.

### Add SIP Trunk

Go to **FreePBX --> SetUp-->Trunks-->Add SIP trunk**

#### Outgoing Settings

Trunk Name:

PEER Details:

```
host=sip.voipbuster.com
secret=password
type=peer
username=aniceman
```

**Trunk Name :** **SIP\_International**

**Peer Details:**

<b>host=sip.voipbuster.com</b>	//the voipbuster SIP server address
<b>secret=password</b>	//your sip account password
<b>type=peer</b>	//can receive and make calls via this trunk
<b>username=aniceman</b>	//your sip account username

#### Registration

Register String:

**Register string: use this string to register the voipbuseter service.**

Format is **username:password@sip serve ip**

## Link outbound route to SIP trunk

Go to **FreePBX --> SetUp-->Outbound Routes-->Add ourbound routes**

Route Name: 8\_international\_calls Rename  
Route Password:   
PIN Set:   
Emergency Dialing: ☐  
Intra Company Route: ☐


### Dial Patterns

8|.

Clean & Remove duplicates

Insert:

### Trunk Sequence

0    
  
Add

**Route name:** 8\_international\_calls  
**Dial Patterns:** 8|.   
**Trunk Sequence:** SIP/SIP\_International

Now we have added a new international route. All number begin with number 8 will be sent to this route.

According to the zap trunk in the above chapter, in our system:

- Calls begin with 8 will outgoing via our SIP service.
- And calls begin with 9 will outgoing via the PSTN line
- Other calls will be regards to internal calls

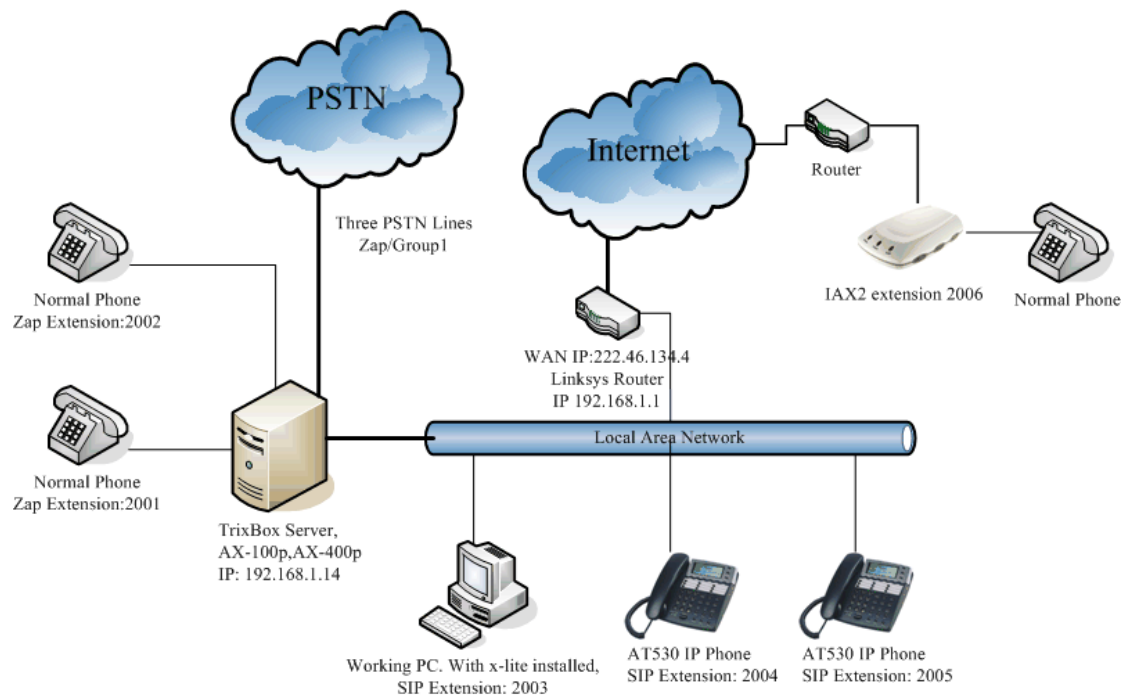
## 6. Remote register through IAX2 protocol

We have built a local simple IP PBX system so far. This is the total structure of my system:

- I lease an ADSL line to connect the internet.
- I have a dynamic WAN IP. (dynamic IP is enough to me and static ip is expensive)
- I have a Trixbox server behind my router.
- I can use my ip phone connect the Trixbox to act as an extension in my local area network. They are at the same network as Trixbox.

And I want to use my Trixbox server when I am on business trip or at home. So how can I do it in this case that my Trixbox server is behind a router and have a private IP? The key point is the **IAX2 protocol** and **port forwarding**.

### Simple structure for small enterprise



Above it the update structure of my system.

I have added a IAX2 extensions 2006 and I use AG-188 ata to register the Trixbox server remotely. Below is the step:

## Add IAX2 extension

Go to -->**Free PBX --> setup-->Extensions -->add generic IAX2 device.** and add a IAX2 extensions:

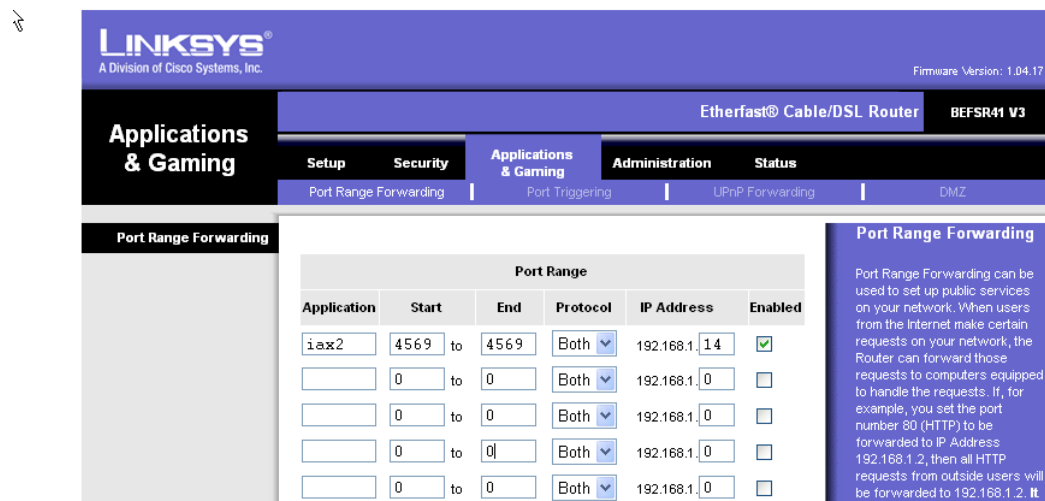
Add IAX2 Exentions:

**User Extension:** 2006 ; Phone number of this extension  
**Display Name:** Alice ; Caller ID  
**Secret:** 2006 ; IAX2 Log on password  
**Enable Voicemail**  
**Voicemail password:** 2006 ; password of your mailbox.

## Add Port Forwarding on router

Since our Trixbox server is behind the router and don't have public ip. We need to do port forwarding in our router so the corresponding packets call be rend to the Trixbox server.

Below is the port forwarding setting in my Linksys router.



The IAX2 protocol use the 4569 port as it register port and voice communicate port. So I forward port 4569 to the public IP. Then all data incoming from the internet via port 4569 will be sent to my Trixbox server (192.168.1.14).

Go to the Linksys status page



**LINKSYS**  
A Division of Cisco Systems, Inc.

Firmware Version: 1.04.17

**Etherfast® Cable/DSL Router BEFSR41 V3**

**Status**

Setup Security Applications & Gaming Administration Status

Router Local Network

**Information**

**Status**

Firmware Version: **1.04.17, Nov 26 2003**

MAC Address: **00-14-BF-AC-7A-F8**

Login Type: **PPPoE**

Login Status: **Connected**

Internet IP Address: **121.35.127.209**

Static DNS1: **202.96.128.86**

Static DNS2: **202.96.134.133**

Static DNS3: **0.0.0.0**

**Router Status**

This screen provides the Router's current status information in a read-only format.

**Login Type**

This field shows the Internet login status. When you choose PPPoE, RAS, PPTP, or HBS as the login method, you can click the **Connect** button to log in. If you click the **Disconnect** button, the Router will not dial up again until you click the **Connect** button.

I can see that my public ip is 124.35.127.209. so I can use my ATA AG-188 to register to my Trixbox server via internet.

## VOIP Gateway

### IAX[Registered] Configuration

IAX Server Addr	<input type="text" value="121.35.127.209"/>
IAX Server Port	<input type="text" value="4569"/>
Account Name	<input type="text" value="2006"/>
Account Password	<input type="text" value="****"/>
Phone Number	<input type="text" value="2006"/>
Local Port	<input type="text" value="4569"/>
Voice mail number	<input type="text" value="0"/>
Voice mail text	<input type="text" value="mail"/>
Echo Test number	<input type="text" value="1"/>
Echo Test text	<input type="text" value="echo"/>
Refresh Time	<input type="text" value="60"/> Seconds
<input checked="" type="checkbox"/> Enable Register	<input checked="" type="checkbox"/> Enable G.729
<input checked="" type="checkbox"/> IAX(Default Protocol)	

The configure method of AG-188 is the same as AT-530.

**IAX Server Addr:** 121:35.127.209

**Account Name:** 2006

**Account Password:** 2006

**Phone Number:** 2006

**Enable Register**

**Enable IAX2 as default protocol**

OK, Now!, we can use the AG-188 to register our Trixbox server now, but it is very annoy because our public IP is dynamic and will change after several hours. Fortunately, there is a service called **DDNS**. Via the DDNS, you can bind your dynamic public IP to a fix domain. For example, I am using a free DDNS service from a Chinese company and they give me an account. I run the DDNS client on the Trixbox server. The client will connect to the DDNS server and send my public ip to them .and the server bind the ip with my register domain. So I can use my domain to access the dynamic ip.

## 7. Echo Cancellation

Echo is a big problem in VoIP and the most possibility echo problem in our system is the making outbound calls via an IP Phone. Below is several ways to improve our voice quality.

### Use ztmonitor to trace the echo

Run `ztmonitor -l v` to trace the port1 and you will see:

```
( # = Audio Level   * = Max Audio Hit )
<------(RX)-----> <------(TX)----->
#####*                #####*
```

You can see the RX voice level and TX voice level when you are talking and if we mute one side for example mute the RX PSTN side and there should be total silent in the RX side. In this case, RX level should be 0 and if we are talking on the TX side and see there are some audio on the RX side, it means there are echo in our talking. The first way to reduce the echo is modify the RX and TX gain, we can go to `/etc/asterisk/zapata.conf` and adjust the `rxgain` and `txgain`. Modify and modify up to the echo is acceptable.

Remember you need to go to the asterisk CLI to reload the `chan_zap.so` module after every time you modify the file.

### Use FXOTUNE tool to configure the line impedance

Every country has different PSTN line impedance. If the line impedance doesn't match, then you will experience more or less echo. `Fxotune` utility is a tool to auto configure the line impedance of your PSTN line and set the corresponding echo coefficient for your line.

This tool is installed default in Trixbox. Use the SSH tool to access the Trixbox and run :

```
asterisk -vvvvvgrc
stop now           //before running the fxotune. You need to stop asterisk
fxotune -i 5
```

The system will test the PSTN line impedance and set the corresponding echo coefficient in the `/etc/fxotune.conf` file. This will take about 20 minutes for to do it. Notice, It will test all the FXO ports in your system , if you don't have the PSTN line connect to this port , it will generate wrong data in the `/etc/fxotune.conf` file.

```
fxotune -s         // to apply the setting
```

Run the asterisk and then you will find that the voice is better than before

More info about this utility, please refer [www.voip-info.org](http://www.voip-info.org), you can find more detail of this tool.

## Install the MG2 echo cancellation

If the echo is still worst or you hear some noise at the beginning of your speech, then we can try to use the MG2 echo cancellation. Trixbox doesn't install this soft echo cancellation in default. The soft echo cancellation may give you a good result in Trixbox.

Download the latest 1.2 version zaptel for asterisk official website [www.asterisk.org](http://www.asterisk.org). before install the zaptel driver , we need to install the kernel source of Trixbox. Use SSH to connect the Trixbox server and run:

### **Install Kernel Source:**

```
yum -y install kernel-devel kernel
```

```
yum -y install kernel-smp-devel //
```

There is a bug in the new kernel , to fix it, you need to:

```
cd /usr/src/kernels/2.6.9-34.0.2.EL-smp-i686/include/linux/  
vi spinlock.h
```

search for rw\_lock\_t in that file, and change it to rwlock\_t, otherwise there will be error when compile the zaptel driver

### **Install Zaptel:**

Download the zaptel driver.

```
cd /usr/src  
tar -zxvf zaptel-1.2.17.tar.gz ;  
cd zaptel-1.2.17  
vi zconfig.h and locate  
/* #define ECHO_CAN_MARK2 */  
/* #define ECHO_CAN_MARK3 */  
#define ECHO_CAN_KB1  
+/* MG2 is a version of KB1 that has some changes to it that are  
+ * supposed to improve how it performs. If you have echo problems,  
+ * try it out! */  
+/* #define ECHO_CAN_MG2 */
```

**Uncomment the** +/\* #define ECHO\_CAN\_MG2 \*/

**And comment** #define ECHO\_CAN\_KB1

```
make clean;make install ; finish the install of zaptel
```

Then MG2 echo cancellation is successful installed in your system.

## Octware echo cancellation

Octasic provides a software echo cancellation for Asterisk, it can improve the voice quality much. Trixbox 2.2 has installed this software (OCTWARE) default. You will need the license key to use it. You can find it easily on the internet, the price is about several US dollars. This is the best choice to improve the voice ☺

## 8. Install AX-4S in the Trixbox

AX-4S is a four ports BRI card. It doesn't use the zaptel driver so the genzapconf command doesn't work on it. If you need to install this card on the Trixbox server, you need to install the mISDN driver, it base on the mISDN driver.

### Configure the jumper and switch of the card

First according to the AX-4S manual to set the card to correct jumper, I use the card to connect the NT Plus's S/T port, so I configure the card's four ports as TE port.

The Jumper setting should be: jumper S401~404 set to right side. Switch 401~402 set to ON(Left side)

And then put the AX-4S card on your PC.

### Install the mISDN driver

Trixbox doesn't have the linux kernel source in their default install. You need to install the kernel source manually otherwise you will have trouble to install the oterh software.

- 1: **yum -y install kernel-devel kernel**
- 2: **yum -y install kernel-smp-devel** // install kernel source
- 3: **cd /usr/src**
- 4: **wget beronet.com/download/install-misdn-mqueue.tar.gz** // get mISDN and chan\_misdn.
- 5: **tar xzfv install-misdn-mqueue.tar.gz**
- 6: **cd install-misdn-mqueue**
- 7: **make**
- 8: **make install**
- 9: **reboot**

After reboot, we have successfully install the misdn driver of the AX-4S and we can use the card now.

### Modify misdn-init.conf and misdn.conf file

The job of /etc/misdn-init.conf file is the same as the zaptel.conf.

I put below on this file:

```
card=1,0x4 ; this is for the AX-4S card  
te_ptmp=1,2,3,4 ; configure port 1~4 to TE port.
```

/etc/asterisk/misdn.conf is the interface of the card and asterisk

Most important is

**[intern]**

```
; define your ports, e.g. 1,2 (depends on mISDN-driver loading order)
ports=1,2,3,4
context = from-pstn
; context where to go to when incoming Call on one of the above ports
msns=*
```

In the misdn.conf file, I configure the port 1~4 as a group named “intern”. to use these ports to make outgoing call. We need to add a custom trunk.

## Add custom trunk:

Go to **FreePBX-->Set up --> Trunks** and add a new custom trunk.  
Configure the outgoing setting as:

### Outgoing Settings

Custom Dial String:

You can also use mISDN/1/\$OUTNUM\$/b to specify the outgoing call via port 1.  
Then add a new route to this trunk and you can make outgoing call via the AX-4S card.

## Add inbound route:

According to the misdn.conf file, our incoming call will go to from-pstn context s extension.

We need to add a route how it should work for this s extensions.

Go to **FreePBX --> SetUp --> Inbound Routes--> Add Inbound Route**

### Edit Incoming Route

DID Number:   
Caller ID Number:

### Set Destination

☐ Recordings:    
☐ Core:    
☒ IVR:    
☐ Custom App:

After doing that, all calls to the BRI port will route to the IVR entry.

## Auto load the AX-4S card after system startup

Open the **/etc/rc.d/rc.local** file and add a new line on this file to auto start the AX-4S card after system startup

```
/etc/rc.d/rc.local
=====
#!/bin/sh
#
# This script will be executed *after* all the other init scripts.
# You can put your own initialization stuff in here if you don't
# want to do the full Sys V style init stuff.

touch /var/lock/subsys/local
/etc/init.d/misdn-init start      <== add this line
/etc/trixbox/runonce
/usr/local/sbin/motd.sh > /etc/motd
/usr/sbin/xfxotune -s
/usr/sbin/amportal start
=====
```

## 9. The End

Thanks for your reading of this article. If you have question or advise of this article, please feel free to contact me at [edwin@atcom.com.cn](mailto:edwin@atcom.com.cn). Next update: Value add service, calling group, FAX, Skype ... and so on